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PROVISIONAL APPLICATION FOR PATENT COVER SHEET

This is a request for filing a PROVISIONAL APPLICATION FOR PATENT under 37 CFR 1.53 (c).

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TITLE OF THE INVENTION (500 characters max)

Quality Determination for Packetized Voice or Video Information

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ENCLOSED APPLICATION PARTS ***(check all that apply)***

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Respectfully submitted,

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Docket Number: PCI-001P

USE ONLY FOR FILING A PROVISIONAL APPLICATION FOR PATENT

Docket No. PCI-001P

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

PROVISIONAL PATENT APPLICATION FOR:

QUALITY DETERMINATION FOR PACKETIZED VOICE OR VIDEO INFORMATION

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QUALITY DETERMINATION FOR PACKETIZED VOICE OR VIDEO INFORMATION

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SPECIFICATION

Introductory Remarks

The pages that follow describe certain embodiments consistent with the above-entitled invention. The description embodied within the following pages is in the form of a "white paper" and other information developed for the purpose of describing methods, devices, systems and services consistent with certain exemplary embodiments of the present invention and provide a level of detail adequate to permit software and hardware designers to make and use an implementation of certain embodiments of the invention. Thus, although this format does not conform to the format of a conventional utility patent application, it conveys all the information needed for those of ordinary skill in the art to make and use one or more embodiments of the present inventions without undue experimentation. This description incorporates many details and specifications that are not intended to limit the scope of protection of any utility patent application which might be filed in the future based upon this provisional application. Rather, it is intended to describe illustrative examples and operational methods associated with such examples. Those skilled in the art will appreciate the many advantages and variations possible on consideration of the following description.

Again, the reader should understand that the present document, while describing specific commercial embodiments and other technical details, should not be considered limiting since many variations of the inventions disclosed herein will become evident in light of this discussion. While this invention is susceptible of embodiment in many different forms, there is shown in the drawings and will herein be described in detail specific embodiments, with the understanding that the present disclosure is to be considered as an example of the principles of the invention and not intended to limit the invention to the specific embodiments shown and described.

Introduction

Historically voice calls have been made using the Public Switched Telephone Network (PSTN). This networking environment has been developed over the past hundred years using technologies that have centered on making telephone companies more efficient through better use of existing wire and new fiber optic facilities. With data usage and the advances in packet technology, Internet Protocol Telephony (IPT) is set to become the preferred networking method thus replacing traditional telephone environments.

The driving factors are compelling for both the economic and application value it brings to service providers, businesses and consumers. From a service provider viewpoint, IPT significantly reduces infrastructure and operational costs. These savings may be passed on to the customer and help the provider improve return on investment. From the customers perspective lower costs are an advantage but more compelling is the possibility of data and voice integration applications that were just not possible with traditional telephony.

Given the value of service provider and customer migration to IPT, it is not surprising that research studies confirm there is pent up demand to transition to IPT. A key assumption supporting this demand is that the fundamentals of reliability and voice quality are at least consistent with, if not better than, the traditional telephone network. Customers will only move if the service levels and voice quality of IPT meet these standards. The traditional providers have set a very high bar for uptime and voice quality, consumers expect close to perfection. The challenge to the IPT service provider is to meet the bar that was set by matching voice quality and service levels then raise it through enhanced application service offerings.

The current state of IPT testing is focused on network and carrier testing but wholly inadequate for measuring the customer experience.

Traditional Telephony

Telecommunications networks have gone through an evolution that has taken a hundred years. This hundred year evolution has created a culture centered on a connection based network and a service provider centric view of network management.

Initially all calls were carried on a single dedicated wire that was connected or cross connected by an operator on a switchboard. Once the connection was made, the callers talked on a pair of wires transporting analog signals from end to end. In this environment, degradation in call quality was determined by signal loss caused by distance and the number of connections frequently resulting in low volume and static. The voice quality solution in this environment was, in part, analog signal regeneration and movement to automated switching centers.

As time and acceptance of the telephone progressed it became impractical to continue stringing new phone wires to meet demand. This was solved by the development of technology that allowed more than one phone conversation to be held simultaneously on a pair of wires creating a multiplexed environment. First analog frequencies were split using frequency division multiplexing (FDM) and multiple calls over a single wire became possible. Using this method, voice quality was somewhat degraded and alas demand outstripped the capacity and again a new method was needed.

The concept developed to resolve both these problems was the introduction of digital signal technology. This technology converted the analog spoken word into digital signals allowing high quality transmission of multiple phone calls over a single transport facility. This technology of pulse code modulation (PCM) was the first introduction of digital technology requiring special testing techniques to measure voice quality. The signal conversion occurred in a coder, de-coder (CODEC). Voice quality testing was accomplished simply by converting

signals back to analog using the same CODEC and measuring the analog signal in the same manner the caller would hear the call, simple but effective.

This brief history of the migration from analog to digital in the telephone network set the stage for a continuing need to gain more and more efficiency in telephone company networks and the requirement to change testing techniques to adapt to new technology and customer demand for quality.

As technology advanced, additional efficiencies were gained through the introduction of embedded signals in the call that caused the network to take action unrelated to the voice conversation being held. This introduction of packet technology created a revolution in the telephone company's core network. The delivery of high quality analog signals was then relegated to the "last mile" (from the phone company's switching center to end users).

In the 1990's customers began to adopt data applications at tremendous rates and the same capacity strains experienced with early telephone deployments and a lack of facilities began to appear in the last mile network. This problem became severe in the late 1990's and was largely solved by the introduction of competitive network builds in the local and long haul facilities. These network builds again primarily incorporated traditional systems.

In every instance traditional telephony has been able to replicate what the customer is hearing. This has been a cornerstone in maintaining the high level of service that customers expect today.

Demarcation of Responsibilities

Traditional telephony is managed by shared responsibility between the carrier and the customer. This shared responsibility requires a clear line be drawn between the two parties. The traditional phone company will only accept responsibility to the last point where they have test capabilities from their central offices, beyond that point it is the customer responsibility to maintain and manage the environment. The result is a demarcation point that determines how the user handles troubles or problems.

Demarcation came into being when the Bell monopoly was forced to allow non-telephone company supplied equipment on customer premises. This resulted after the federal courts ruled in the benchmark Carterphone decision in 1974. Since that time phone companies have used the demarcation as a cornerstone for how mixed vendor environments are managed.

This key event is also the reason so much innovation has occurred in telecommunications in the last 30 years. It provided the competitive incentive for the Bell companies and outside suppliers to invest time and money. The greatest innovation may now be upon us with the introduction of IPT.

Implementation of IPT by its nature grays the demarcation point. This is very disruptive to the traditional phone companies. How providers manage the elimination of the demarcation point will be a large part of determining their success in the market.

Introduction of the Internet and the World Wide Web (WWW)

In the mid 1990's companies began to embrace the Internet and its core protocol (IP) as a preferred method to access data information. The convenience, availability, applications and relatively low cost created compelling reasons to begin using IP and the WWW as the data transport methodology of choice.

This environment uses packet technology to transport and deliver information between networks and individual desktop computers. It is highly efficient in the way it takes advantage of associated but separate computer networks to accomplish its mission of delivering information. Using voice transport technologies initiated by telephony needs over this data network has presented both opportunities and problems. It has become possible to transport phone calls over IP and further to have voice and data applications interact because of the use of a common protocol. The problems come from two primary areas; the differences in the ways networks are managed and the way in which voice quality is determined.

Any technology that replaces traditional telephony transport and delivery must replicate the tried and true quality barometer, the human ear, as the primary method of determining voice quality. To date, quality assurance testing of IP Telephony has been data centric. Monitoring and testing methods are drastically different between voice and data; therefore new methods of testing IPT, that are voice centric, must be explored.

Internet Protocol Telephony

IPT and traditional telephony differ in many ways but have a common goal of delivering voice communication in real time between two or more parties. The nature of IP is such that this delivery can be affected by variables different than those that affect traditional phone calls. The biggest difference with IPT is analog voice information can be converted into a data packet immediately (<u>as early as within the phone itself</u>) and can remain as data until it reaches the phone at the other end. So as you can see, measuring analog voice quality, as it is perceived by the caller, is fundamentally different than traditional telephony.

Similar to PCM as outlined above, IPT relies on CODEC technology to convert information from analog to digital signals and compress the information. But unlike PCM, the CODEC adds network-signaling information at the customer premise - the end user desktop in fact. Add to this a high dependence on the customer local area network (LAN) and its equipment infrastructure what you end up with is an extension of a network once the sole domain of the service provider or phone company that now includes the customer's own LAN and all its vagaries.

Causes of Poor Voice Quality

The causes of poor voice quality can be attributed to almost any piece of network equipment that acts upon the voice packet information or the transport network itself. IPT is highly dependent on logical transport and route management where traditional telephony is generally affected more by physical transport management. The result of these different transport management methods is that IPT voice quality management is far more critical than traditional telephony.

IPT uses statistical management to determine voice packet forwarding and routing. By its nature statistical management makes decisions on which packets go at what time and to whom. During times of high traffic load, the packet processors need to make critical decisions on what voice information to send and when to send it. This means that inevitably some packets will be dropped or lost. As stated earlier, this is not critical for data, data can be resent; but to drop information in the middle of a voice call, information cannot be resent, producing the "chopping" or "clipping" effect. Poor congestion management can result in poor voice quality.

Errors occurring on the physical transport facility can cause the same effect as congestion. If an error is taken on a physical transport facility the entire voice packet is lost. In traditional telephony most transport will be unaffected by low error rates on the physical transport facilities.

In all the router and gateway environments that make up pieces of an IPT network, decisions must be made giving priority of some packet information over others. If prioritization is not optimized for voice traffic, delay can occur producing loss of packets jitter or latency as a result and thus, poor quality.

The ability of the network equipment to process packet information timely is crucial to moving packets quickly to the next destination. Processor load can cause delay in forwarding packets and also result in jitter, latency or packet loss.

In most packet networking equipment, information is stored in memory prior to forwarding to the processor. If memory becomes full or overloaded, packets will be dropped causing voice packet loss.

Managing the Caller Experience

Fundamentally, caller experience is the same with IPT as with traditional telephony. Because the ear is an analog hearing device, the most important point of measurement is where the digital information is converted to analog. But unlike traditional telephony, it is either impossible or impractical to "plug in" a testing device to determine the quality of a call. Therefore, the digital information should be measured in a way that mimics what the human ear processes.

In order to ensure you are testing and monitoring the actual customer experience, monitoring CODEC to CODEC may be the most (and possibly only) valid testing point. The challenge comes because it is impractical to install dedicated testing systems at each CODEC or customer phone. Thus, to resolve this issue, testing methods should be developed without dedicated equipment at each CODEC end point, but that still produce CODEC to CODEC test results.

Perceptual measurement techniques such as PSQM and PESQ measure the difference between a reference analog signal and a degraded analog output. These techniques use a known reference, usually a standardized recorded phrase in order to accurately measure a call. They generally use a controlled environment and outboard testing systems. Testing using PSQM and PESQ have become preferred methods during network setup and general network failures. Although accurate, these methods are impractical at customer locations due to the cost of testing systems and the intrusive nature of the test. These are best suited for use within the service provider network.

Many service providers attempt to avoid the potential problems of voice quality by overbuilding the transport and network components. This environment, although initially effective, only masks the potential long-term problem and reduces the value of IPT because network efficiency is not maximized. With the current abundance of capacity this method is initially attractive but does not effectively prepare the network environment for the inevitable need to maximize utilization.

Some providers and users resort to reactive management based on customer complaint. Once a caller complains, an engineer or technician can draw statistics from different network elements and deduce suspected causes. Then, largely by trial and error, corrections are made to the network. This is highly undesirable in an environment where callers expect perfection on every call.

PingTone's CADIUS™ Approach

PingTone Communications, Inc. has developed a new approach to IPT call quality management; the CADIUS System. With the goal of measuring actual calls from end to end, CADIUS is effective in assuring a caller experience on par with traditional telephony.

The CADIUS System results in the following environment:

-Every call is measured
-Testing is CODEC to CODEC, Caller to Caller
-The actual caller experience is evaluated and reported
-Call quality events & problems are isolated to specific network sections and components
-Management information is presented in near real time

The most significant advancement in CADIUS is that actual calls are used to evaluate the caller experience. Information is derived from the packet stream of the call and applied to an algorithm that assigns a score to the call.

If the call's score is outside predetermined acceptable limits, proactive measures can be taken to improve call quality. In most cases the caller will not perceive the deviation in quality that would cause proactive measures to be taken.

PingTone's approach pulls information from the real-time transport protocol (RTP) information of the voice packet and applies a calibrated formula to determine a quality score of the call. By making calculations from embedded data within the IP packet, tests are completely non-intrusive and passive.

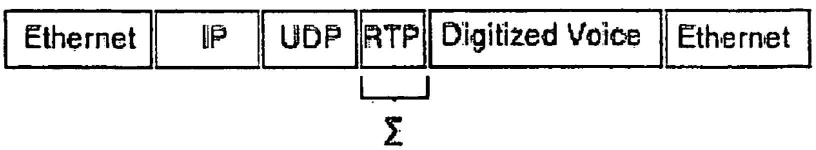

Figure 4

In one embodiment consistent with the present invention, the GeoPacket Call Quality Analyzer (CQA) is software that provides a real-time view of conditions affecting voice quality on a Voice-over-IP (VoIP) network. The software resides on one or more computer systems on the network, and collects data from one or more sources, including (but not limited to):

- Samples of network communications, in which digitized voice media is typically carried in Real Time Protocol (RTP) packets

- Metrics contained within Real Time Control Protocol (RTCP) packets

- Service Assurance Agent probe results recorded on a Cisco VoIP gateway

- Call metrics stored on a softswitch or other VoIP network element

The CQA is not an active element of the VoIP network; that is, it does not participate in signaling, or otherwise directly interfere with the setup, progress, or teardown of VoIP sessions. It may, however, provide input to other VoIP network elements that may affect VoIP sessions based on the input (for example, by rerouting calls to higher-quality links).

From the data collected by the CQA, the values of basic network metrics such as jitter, packet loss, and latency are determined for ongoing VoIP sessions and/or for links between elements of the VoIP network. These values are then inserted into a formula that generates a numerical score characterizing the fidelity of vocal communications during the session or carried over the link. Scores are generated frequently and made available for display (for instance, from a web server) or further processing.

The formula is not necessarily a steady state formula. It is determined by calibration of each deployment and varies based upon the equipment configuration primarily, but not exclusively, based on the IP device (i.e., the IP telephone or IP video appliance, etc.).

To determine the formula, independent measurements of voice quality on the subject VoIP network are made. These measurements may be subjective (e.g., ratings by human listeners) and/or objective (e.g., scores produced by computerized voice quality testers). Simultaneous samples of network metrics (jitter, packet loss, latency, etc.) are taken by the CQA for the same sessions or links being independently observed and rated. This results in network metrics correlated to the voice quality scores resulting from the independent measurements.

With sets of correlated data in hand, relationships between the independently measured quality scores and the network metrics are studied in order to determine a function of the available network metrics that best matches the output of the independent measurements. Ideally, quality measurements may be taken in which all but one of the correlated metrics are relatively constant, permitting study of the relationship between voice quality and the variable metric in isolation, but this is not often the case. A variety of techniques may be used to produce a formula from the correlated data that offers results of the desired accuracy and precision: linear regression analysis, curve fitting, graphing, etc. Since the relationships between quality scores and network metrics are generally multi-variate and non-linear, and since the data source used by the CQA may not provide all the relevant data required to fully characterize voice quality, a series of judicious guesses may be tried to determine the formula with an optimal fit to empirical measurements.

Once the basic formula has been calibrated against independent measurements of voice quality, the range and distribution of quality scores generated by the formula may be modified to correspond to any commonly used voice scoring system (such as MOS, PSQM, PESQ, MNB, or R factor), so that the meaning of the numerical scores will be evident to those familiar with the scoring system. In order to keep the relationship to empirical measurements intact, care should be taken during the conversion to maintain numerical correlations between checkpoints in the source and destination scoring systems (that is, scores in the two systems that are recognized as applying to similar conditions).

As mentioned earlier, while monitoring quantities such as latency, jitter, and packet loss in isolation does not offer a service provider an accurate view of voice quality on a network, a combination of these metrics in an algorithm calibrated against independent measures of quality can deliver real-time results reflecting caller experience. The same method may be used to generate quality scores for non-voice audio, video, fax, or other forms of telecommunication over a packet-switched network. Many other embodiments are possible within the scope of one of ordinary skill in the art, in view of the present teaching.

Thus, the process takes real time samples of embedded packets and applies an algorithm which emulates the analog qualities as experienced by the human ear or eye (listener or viewer). This information can then be used to manage and troubleshoot data infrastructures being used for voice and video applications.

Conclusions

If one believes that in order to realize the true value of IPT, it is best to deliver a full IPT implementation as close to the caller and desktop computer as possible, then one should also acknowledge that the service provider network now includes network elements both within their controlled environment and outside of it effectively eliminating the demarcation point. Customer local area network (LAN) components then, such as routers, switches, hubs and firewalls can have a deleterious effect on voice quality, so much so that measuring and testing calls to include these components should become a service component.

In order to ensure that the full value of IPT is realized, service providers should accept that measurement of the entire calling experience should be captured. Until this happens, the value of IPT most likely will not go beyond providing lower cost basic telephone services.

The demarcation culture, approaches and methodology derived from traditional telephony that measure quality only within the service provider network will most likely result in customer dissatisfaction with the IPT experience. Service providers who embrace IPT may need to break old traditions and create new methods of delivering assured voice quality that are at least on par with traditional telephony.

The reader should understand that the present document, while describing specific features of one embodiment consistent with the present invention, should not be considered limiting since many variations of the inventions disclosed herein will become evident in light of this discussion. The invention itself is susceptible of embodiment in many different forms. The present disclosure is to be considered as an example of the principles of the invention and not intended to limit the invention to the specific embodiments shown and described.

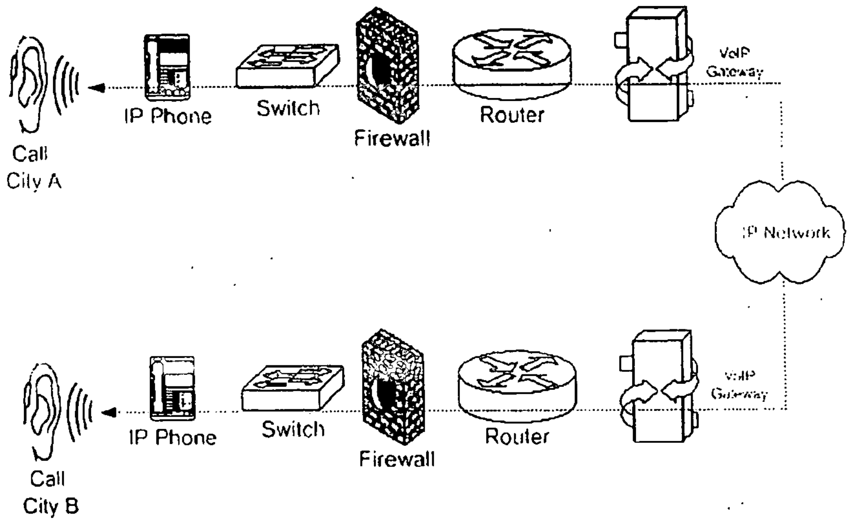

Figure 1

Factors that affect IPT voice quality

Keeping in mind that IP was originally designed and optimized for data traffic, its characteristics are not necessarily ideal for providing high quality voice. In the data environment information is generally not time sensitive, has little concern for lost bits of the information stream (as it can be retransmitted) and is not particularly affected by variations in the packet-to-packet delivery timing. Voice, on the other hand, can be affected by any or all of these issues.

Latency is the time it takes for a voice packet to leave the origination and arrive at the destination. High latency with respect to voice usually results in an echo effect.

Packet Loss is simply information that is sent from point A and is not delivered to the intended destination. This can cause an unintended clipping, choppiness or silence during a call.

Jitter is the variation in the delay time from one packet to the next. In traditional telephony, all network components are carefully timed by a master clock keeping each piece of information in strict time sequence. With IPT, gateways, routers, switches and firewalls make mostly individual decisions as to when to forward voice packets. This can result in this variation in packet-to-packet delivery time causing the call to have a "worble" effect.

All is not lost however, as it is relatively easy to adjust network equipment to ensure optimization for voice. The true difficulty lies in identifying when and where a problem exists and what factor or factors are causing the voice quality degradation.

CODECs used in IPT are actually quite adept at adjusting to most quality impacting events. Where they have pronounced difficulty is "understanding" the effect on the listener when one or more quality degrading events occurs. Simply monitoring latency, packet loss and jitter will not tell the service provider what the caller is experiencing.

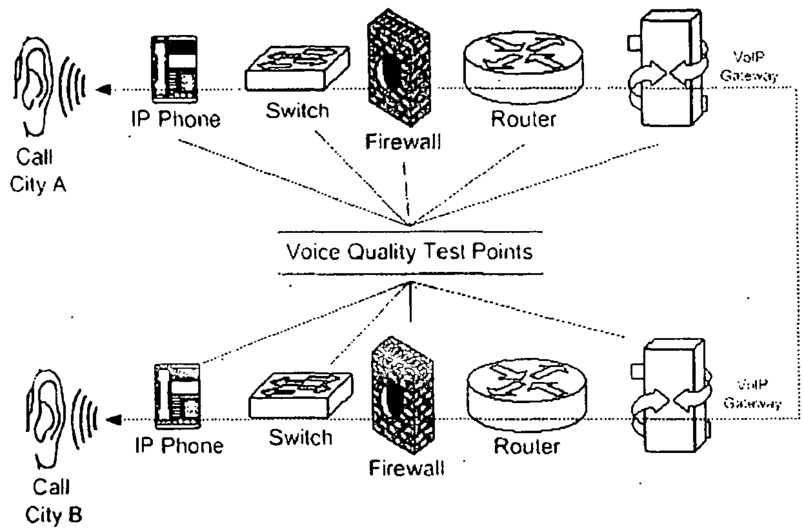

Figure 2

With end-to-end IPT it is possible and generally most desirable to deliver the pure IP call information as close to the caller as possible. This allows for deep voice and data integration to the user phone and desktop computer. Certain partial IPT implementations convert digital to analog near or at the edge of the customer network rather than at the end user desktop. This is done either within the service provider network or at the customer private branch exchange (PBX) in order to lower customer usage costs. A partial implementation, however, makes it impossible to get the optimal value in IPT.

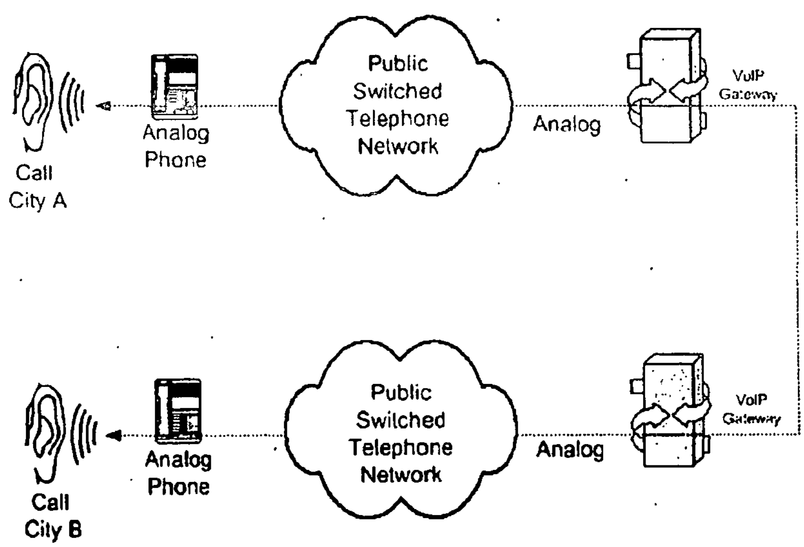

Figure 3